# Omnia ONE

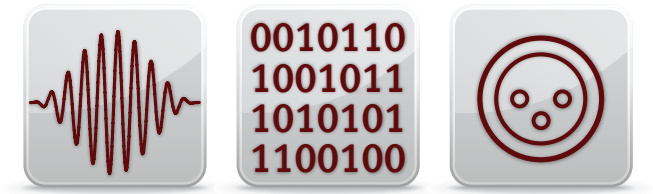
STICKER SHOCK. THE GOOD KIND.

5,000+ Omnia ONE's. That's how many are on the air in your area and around the world. Engineers tell us of reliablilty and ease of operation. Program Directors love the power and punch of genuine Omnia processing. General Managers appreciate that Omnia ONE can become an FM, AM, HD Radio, DAB. DRM, multicasting, podcasting, netcasting or satcasting unit as needed.

All of them love the price.

## Universal Advantages

Choose software downloads for FM, AM, internet/satellite or studio processing on demand if necessary




Balanced XLR Analog inputs and outputs, plus Digital AES/EBU input, output and external Sync

Browser-based remote control and configuration

Automatic backup input switching

# Universal Advantages (Cont.)

Livewire / Ethernet RJ45 jack

Universal Power Input

Robust headphone amp with rugged jack and front panel volume control

Single Jog-Wheel and Selection Button user interface with LED level metering and LCD screen

## FM Advantages:

Smart wideband AGC followed by advanced Four-Band AGC and selectable Four or Five-Band Peak Limiter sections. Omnia's advanced, fully distortion-controlled, pre-emphasized final limiter / clipper. A newly designed digital stereo generator with SCA convenience input, two independently adjustable composite MPX outputs and 19kHz pilot output for synchronization to external RDS generators.

## AM Advantages:

Omnia's advanced NRSC compliant, distortion-managed final limiter / clipper, including selectable Low Pass Filter frequencies that support AM HD transmission installations...the same as used in the Omnia ONE's bigger siblings.

## Omnia ONE Multicast Advantages

Features SENSUS™, an audio conditioning technology to minimize codec artifacts as well as restore the fullness and depth that bit-reduction steals. See expanded explanation about SENSUS technology later in this brochure. Ultra low-distortion final limiting optimized for the HD codec.

## Omnia Studio Pro Advantages:

Full-bandwidth processor for applications that require minimal delay and do not require absolute peak limiting.
The first studio processor to include a four-band compressor / limiter allowing you precise and accurately defined control while pre-processing music, commercials, remote feeds, or sweetening audio. Applications include recording studios, mastering labs, TV stations, radio headphone feeds ... just abut any application where signal processing is needed.

# Omnia ONE Overview

# Omnia ONE Physical Specifications

## Dimensions:

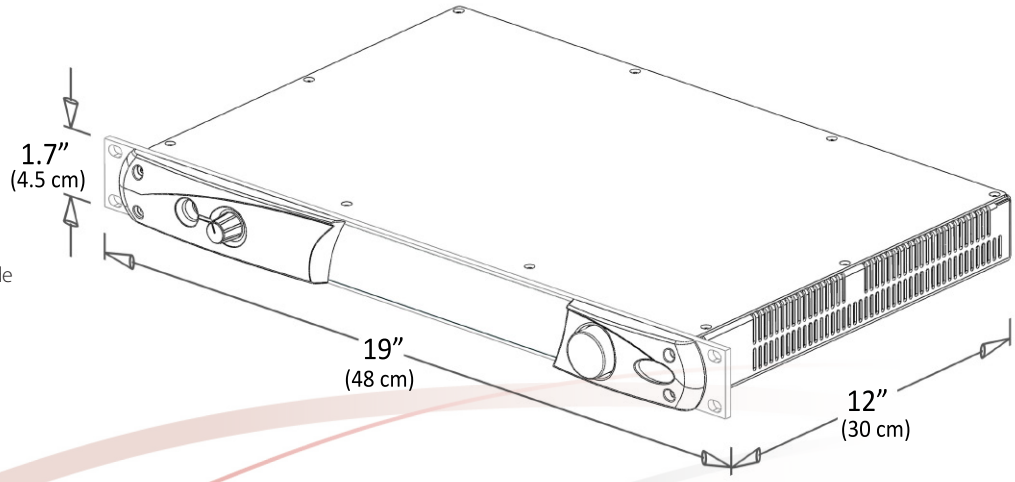
- 19" wide x 1.7" high x 12" deep (48 cm wide x 4.5 cm high x 30 cm deep). Unit requires one EIA rack spaces for mounting.
- Allow up to 4" in rear for connector and cable clearence.
- 6 lbs. unit weight

## Shipping Weight:

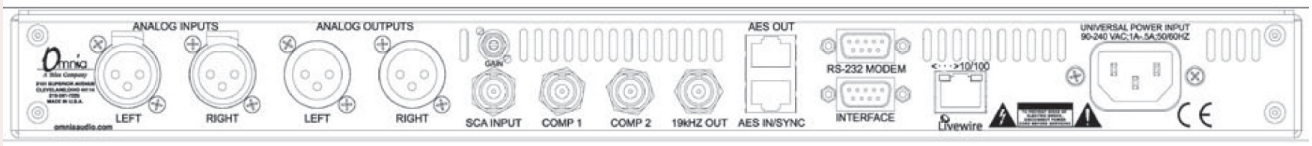
- 12 lbs. / 5.5 kg

## Accessories:

- Omnia ONE Manual
- Omnia ONE Quick Start Guide
- Warranty Registraton
- 120V AC power cord
- 240V AC (Euro power cord)
- Rack screws (bag of five)




# Omnia ONE Back Panel



# Omnia ONE Performance Specifications

## Omnia ONE FM Specs.

### Frequency Response:

- Complies with the standard 50 or 75 microsecond pre-emphasis curve within ± 0.50 dB, 30 Hz to 15 kHz. The analog left/right outputs and AES/EBU Digital outputs can be configured for flat or pre-emphasized output..

### System Distortion:

- Less than 0.01% THD, 20 Hz – 7.5 kHz. Second harmonic distortion above 7.5 kHz is not audible in the FM system.
- ***Signal-Noise Ratio:** > -80 dB de-emphasized, 20 Hz –- 15 kHz bandwidth, referenced to 100% modulation).

*The measured noise floor will depend upon the settings of the Input and Output Gain controls and is primarily governed by dynamic range of the Crystal Semiconductor CS5361 A/D Converter which is specified as >110 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

### Stereo Separation:

- Greater than 65 dB, 20 Hz –- 15 kHz; 70 dB typical.

### Crosstalk:

- > -70 dB, 20 Hz -- 15 kHz.

### Composite Outputs:

- **Source Impedance:** 5 ohms or 75 ohms, jumper-selectable. Single-ended and floating over chassis ground.
- **Output Level:** 0V to 10V in 0.05V steps, software adjustable.
- **D/A Conversion:** Texas Instruments/Burr Brown PCM1798, 24-bit sigmadelta converter.
- **Configuration:** Two electrically independent outputs. Software based level adjustment.
- Load Impedance: 50 ohms or greater load is suggested.
- **Pilot Level:** Adjustable from 4.0% to 12.0% in 0.1% steps and OFF.
- **Pilot Stability:** 19 kHz, ± 0.5 Hz.
- **Signal-to-Noise Ratio:** -85 dB typical, 75 µS de-emphasized, 15 kHz bandwidth, referenced to 100% modulation).
- **Distortion:** < 0.02% THD 20 Hz – 15 kHz bandwidth, 75 µS deemphasized, referenced to 100% modulation.
- **Stereo Separation:** > 65 dB, 30 Hz – 15 kHz.
- **Linear Crosstalk:** > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).
- **Non-linear Crosstalk:** > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).
- **38 kHz Suppression:** > 70 dB (referenced to 100% modulation).
- **76 kHz Suppression:** > 80 dB (referenced to 100% modulation).
- **Pilot Protection:** > -65 dB relative to 9% pilot injection, ± 1 kHz.
- **57 kHz (RDS/RBDS) Protection:** better than -50 dB.
- **Connectors:** Two EMI suppressed female BNC, floating over chassis ground.
  - **Maximum Load Capacitance:** 5nF (at 10 ohms source impedance).
  - **Maximum cable length:** 100 feet/30 meters RG-58A/U.

### Analog Audio Input:

- Left/Right Stereo.
- Electronically balanced.
- Input impedance 10k ohms resistive.
- **Maximum Input Level:** +22 dBu.
- **Nominal Input Level:** +4dBu, which nets a -18dBFS input meter reading on a steady-state signal when the Input Gain control is set to 0.0dB. Program material with a nominal average level (VU reading) of +4dBu will typically produce peak readings on the input meter in the range of -12 dBFS to -6dBFS. This is the correct operating level.
- **A/D Conversion:** Crystal Semiconductor CS5361, 24 bit 128x oversampled delta sigma converter with linear-phase anti-aliasing filter. Pre-ADC anti-alias filter, with high-pass filter at <10 Hz.
- **Connectors:** Two, EMI-suppressed XLR-female. Pin 1 chassis ground, Pin 2 "Hot".

### Analog Audio Output:

- Left/Right Stereo. Electronically balanced.
- Output Impedance 20 ohms.
- **Minimum load Impedance:** 600 ohms.
- Output Level adjustable from -2 dBu to +22dBu peak in 0.1dB steps.

### D/A Conversion:

- Crystal Semiconductor CS4391, 24 bit, 128x oversampled.
- **Connectors:** Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot".

# Omnia ONE Performance Specifications (Cont.)

## Digital Audio Input:

- **Configuration:** Stereo per AES/EBU standard, CS8420 Digital Audio Transceiver with 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum. Automatically accepts and locks to input sample rates between 32 and 108 kHz.
- **Connector:** EMI-suppressed RJ-45 female pinned according to StudioHub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Audio Output:

- Stereo per AES3 standard. Digital Output sample rate can lock to the input, lock to an additional external sync source, or use the internal 48kHz rate.
- **Connector:** EMI-suppressed RJ-45 female according to Studio-Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Output Level:

- -24.0 to 0.0 dBFS peak, software adjustable in 0.1dB steps.

## External Sync Input:

- **External Sync:** Allows the output sample rate to be synchronized to an AES3 signal applied to the Ext. Sync input connector. (Does not accept Word Clock inputs)
- **Connector:** EMI-suppressed RJ-45 female according to Studio-Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## External Sync Range:

- Automatically accepts sample rates between 32kHz and 96kHz.
- **Connector:** EMI-suppressed RJ-45 female pinned according to StudioHub+® standards, Transformer isolated, balanced, and floating according to AES3 standard.

## Remote Control Methods:

- External Modem or 10/100BaseTX Ethernet.
- **--Modem (support not yet available as of software version 2.2):** User-supplied Hayes command compatible external serial modem connected to rear-panel DB-9 male serial port.
- **-- Ethernet:** TCP/IP control via web page interface and Java (TM) remote control program included in the web pages. All software is served from the built-in web server; there is nothing to install on the user's computer.
- **Ports Used:** The defaults are TCP Ports 4545 and 4546 (for control and metering data, respectively).
- **Connectors:** Modem port - EMI-suppressed DB-9 male connector. Ethernet - Industry standard EMI-suppressed RJ-45 connector.

## GPI Interface:

- Support not yet available as of software version 2.2.
- **Connector:** EMI suppressed DB-9 female connector.

## Power Requirements:

- **Voltage:** 100-250 VAC, 47-63 Hz., Less than 40 VA.
- **Power Connector:** EMI suppressed IEC male. Detachable 3-wire power cords supplied for US and European use.
- **Power Supply:** Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.
- **Environmental:** Operating Temperature: 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges. Humidity: 0-95% RH, non-condensing.

# Omnia ONE AM Specs.

## Note:

- Discrete I/O measurements have been made in "Bypass" mode (available in the Input/Output menu).

## Note:

- All measurements made with the supplied "FACT_TEST" preset, which is available in the Preset Submenu.

## System Frequency Response:

- Complies with the NRSC emphasis curve within ± 0.50 dB, 30 Hz to 10 kHz. (At a setting of "10" on the HF EQ control)

## System *Signal to Noise Ratio:

- -80 dB de-emphasized, 20 Hz –- 10 kHz NRSC bandwidth, referenced to 100% modulation).

*The measured noise floor will depend upon the settings of the Input and Output Gain controls and is primarily governed by dynamic range of the Crystal Semiconductor CS5361 A/D Converter which is specified as >110 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

## System Distortion

- Less than 0.01% THD, 20 Hz – 5 kHz. (second order harmonic distortion above 5 kHz is not relevant in the AM system due to the removal of harmonics by the system's 10 kHz low pass filter)

# Omnia ONE Performance Specifications (Cont.)

## System Stereo Separation

- Greater than 65 dB, 20 Hz –- 10 kHz; greater than 70 dB typical.

## Analog Audio Input:

- Left/Right Stereo.
- Electronically balanced.
- Input impedance 10k ohms resistive.
- **Maximum Input Level:** +22 dBu.
- **Nominal Input Level:** +4dBu, which nets a -18dBFS input meter reading on a steady-state signal when the Input Gain control is set to 0.0dB. Program material with a nominal average level (VU reading) of +4dBu will typically produce peak readings on the input meter in the range of -12 dBFS to -6dBFS. This is the correct operating level.

## A/D Conversion:

- Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter. Pre-ADC anti-alias filter, with high-pass filter at <10 Hz.
- **Connectors:** Two, EMI-suppressed XLR-female. Pin 1 chassis ground, Pin 2 "Hot."

## Analog Audio Output:

- Left/Right Stereo. Electronically balanced.
- Output Impedance 20 ohms.
- **Minimum load Impedance:** 600 ohms.
- Output Level adjustable from -2 dBu to +22dBu peak in 0.1dB steps.
- **D/A Conversion:** Crystal Semiconductor CS4391, 24 bit, 128x oversampled.
- **Connectors:** Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot."

## Digital Audio Input:

- **Configuration:** Stereo per AES/EBU standard, CS8420 Digital Audio Transceiver with 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum. Automatically accepts and locks to input sample rates between 32 and 108 kHz.
- **Connector:** EMI-suppressed RJ-45 female pinned according to StudioHub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Audio Output:

- Stereo per AES3 standard. Digital Output sample rate can lock to the input, lock to an additional external sync source, or use the internal 48kHz rate.
- **Connector:** EMI-suppressed RJ-45 female according to Studio-Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Output Level:

- -24.0 to 0.0 dBFS peak, software adjustable in 0.1dB steps.

## External Sync Input:

- **External Sync:** Allows the output sample rate to be synchronized to an AES3 signal applied to the Ext. Sync input connector. (Does not accept Word Clock inputs)
- **Connector:** EMI-suppressed RJ-45 female according to Studio-Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## External Sync Range:

- Automatically accepts sample rates between 32kHz and 96kHz.
- **Connector:** EMI-suppressed RJ-45 female pinned according to StudioHub+® standards, Transformer isolated, balanced, and floating according to AES3 standard.

## Remote Control Methods:

- External Modem or 10/100BaseTX Ethernet.
- **--Modem (support not yet available as of software version 2.2):** Usersupplied Hayes command compatible external serial modem connected to rear-panel DB-9 male serial port.
- **-- Ethernet:** TCP/IP control via web page interface and Java (TM) remote control program included in the web pages. All software is served from the built-in web server; there is nothing to install on the user's computer.
- **Ports Used:** The defaults are TCP Ports 4545 and 4546 (for control and metering data, respectively)
- **Connectors:** Modem port - EMI-suppressed DB-9 male connector. Ethernet - Industry standard EMI-suppressed RJ-45 connector.

## GPI Interface:

- Support not yet available as of software version 2.2.
- **Connector:** EMI suppressed DB-9 female connector.

## Power Requirements:

- **Voltage:** 100-250 VAC, 47-63 Hz., Less than 40 VA.

# Omnia ONE Performance Specifications (Cont.)

- **Power Connector:** EMI suppressed IEC male. Detachable 3-wire power cords supplied for US and European use.
- **Power Supply:** Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.

## Environmental:

- **Operating Temperature:** 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges. Humidity: 0-95% RH, non-condensing.

# Omnia ONE Multicast/DAB & Studio Pro Specs.

## Note:

- All measurements made using "Bypass" mode, which is available in the Input/Output menu.

## General Audio Specifications

- **Frequency Response:** ± 0.50 dB, 20 Hz to 20 kHz with high pass filter disabled.
- **Distortion:** Less than 0.05% THD 20 Hz – 20 kHz bandwidth.
- ***Signal-Noise Ratio:** Greater than -100 dB, 20 Hz –- 20 kHz bandwidth, referenced to 0dBfs

*The measured noise floor will depend upon the settings of the Input and Output Gain controls and is primarily governed by dynamic range of the Crystal Semiconductor A/D Converter which is specified as >100 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

## Stereo Separation:

- Greater than 80 dB, 20 Hz –- 20 kHz; 90 dB typical.

## Analog Audio Input:

- Left/Right Stereo.
- Electronically balanced.
- Input impedance 10k ohms resistive.
- Maximum Input Level +24 dBu.
- Nominal Input Level: +4dBu (A +12dBu input results in –12dBFS input meter reading with Input Gain set to 0.0dB. A 0dBu input signal results in a -12dBFS input level when Input Gain is +12dB.)

## A/D Conversion:

- Crystal Semiconductor 24 bit 128x oversampled delta sigma converter with linear-phase anti-aliasing filter. Pre-ADC anti-alias filter, with highpass filter at <10 Hz.
- **Connectors:** Two, EMI-suppressed XLR-female. Pin 1 chassis ground, Pin 2 "Hot."

## Analog Audio Output:

- Left/Right Stereo. Electronically balanced.
- Output Impedance 20 ohms.
- Minimum load Impedance 600 ohms.
- Output Level adjustable from -2 dBu to +22dBu peak in 0.1dB steps.
- **D/A Conversion:** Crystal Semiconductor CS4390 24 bit, 128x oversampled.
- **Connectors:** Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot."

## Digital Audio Input:

- **Configuration:** Two-channel stereo per AES3 standard via CS8420 Digital Audio Transceiver with 24-bit resolution. Software selection of stereo, mono from left, mono from right or mono from sum. Automatically accepts sample rates between 24 kHz and 96 kHz.
- **Connector:** EMI-suppressed RJ-45 female pinned according to StudioHub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Audio Output:

- Stereo per AES3 standard. Digital Output sample rate software selectable for internal 48kHz, synchronize to AES input, or synchronize to auxiliary AES sync input (per AES-11 / DARS).
- **Connector:** EMI-suppressed RJ-45 female according to Studio-Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Output Level:

- -22.0 to 0.0 dBFS peak, software adjustable in 0.1dB steps.

## Digital Sync Input:

- Output sample rate can be synchronized to the signal present on the AES/EBU input or to the AES3 signal applied to the Ext. Sync connector.

# Omnia ONE Performance Specifications (Cont.)

## External Sync Range:

- Accepts 32kHz to 96 kHz for synchronization of the Digital Output signal to an external reference. Automatically accepts sample rates between 32kHz and 96kHz.
- **Connector:** EMI-suppressed RJ-45 female pinned according to StudioHub+® standards, Transformer isolated, balanced, and floating according to AES3 standard.

## Remote Control Methods:

- External Modem or 10/100BaseTX Ethernet.
- **--Modem (support not yet available as of software version 2.2):** Usersupplied Hayes command compatible external serial modem connected to rear-panel DB-9 male serial port.
- **-- Ethernet:** TCP/IP control via web page interface and Java (TM) remote control program included in the web pages. All software is served from the built-in web server; there is nothing to install on the user's computer.
- **Ports Used:** The defaults are TCP Ports 4545 and 4546 (for control and metering data, respectively).
- **Connectors:** Modem port - EMI-suppressed DB-9 male connector. Ethernet - Industry standard EMI-suppressed RJ-45 connector.

## GPI Interface:

- Support not yet available as of software version 2.2.
- **Connector:** EMI suppressed DB-9 female connector.

## Power Requirements:

- Voltage: 100-250 VAC, 47-63 Hz. Less than 25 VA.

## Power Connector:

- EMI suppressed IEC male.
  Detachable 3-wire power cords supplied for US and European use.

## Power Supply:

- Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.

## Environmental:

- Operating Temperature: 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges. Humidity: 0-95% RH, non-condensing.

# FAQ's

## I can't seem to find a remote control software download for the OmniaONE. Why?

None is needed. Remote control for the OmniaONE is performed over an Ethernet network connection using any standard web browser that can run the Java plug-in.

## How can I convert my OmniaONE to a different style?

Similarly to the Omnia-3 before it, the Omnia One hardware is the same for all styles.
Files are available for free download from the Omnia website to convert the OmniaONE between all of its available styles, currently AM, FM, Multicast and Studio Pro. These are the same files that are used for software updates.

## How much time delay does my Omnia ONE FM have?

The short answer is "Not enough to drive your DJ's crazy!" Seriously, we have measured the delay from the input to any output at approximately 8 ms.

**Tip:** Try both settings of the "Invert" setting in the Input/Output / Input Menu and ask your talent which they prefer. One of the two settings will put what they are hearing through their headphones more in phase with what they are hearing through their own bone conduction. You should find that one of the settings sounds better to them and it should be consistent with all of the talent monitoring off-air using the same feed. This parameter changes the phase of both channels at once, similar to reversing pins 2 & 3 on both channels Input XLR connectors.

After adjusting this setting, most talent gets used to the "new sound" within a few breaks.

**Note:** If there is additional delay added to the air chain, such as from a Digital STL (especially a codec based STL), the cumulative delay may become excessive and cause discomfort for the on-air talent. Real-world tests have determined that a slight echo may begin to be heard at about 15 ms, and anything above 25-30 ms is usually too annoying to talent monitoring off the air.

# FAQ's (Cont.)

### Are all of the output jacks on my Omnia active simultaneously?

Yes, with one exception: Depending on model and style, some physical output jacks may not be active, for example, the BNC composite MPX ouptut jacks on an Omnia ONE will be inactive when a style other than FM is running (AM, Multicast or Studio Pro).

### How do I load one of the other software styles that are available for free download from your website into my Omnia ONE unit?

The procedure is exacly the same as a software update as described in Appendix C of the user manual.

The Omnia ONE has two software banks and when updating, the "other" bank (the bank that is currently not in use) is always used.

So simply perform a software update using the desired style downloaded from the website. To activate the new style, boot into the desired bank from the Configuration tab in the webpage interface.

Note that only one style can be active at a time.

# More Omnia ONE Advantages

### Time alignment.

The Omnia ONE is completely, 100% time-aligned. This means that all audio signals, no matter what frequency, have the exact same propagation time from Input to Output of the Omnia audio processor. This is a claim that other manufacturers cannot make, as they don't deem time-alignment to be important. Omnia processors sound more precise and less "smeary" due to this attention to time-alignment.

### Pre-emphasis placement.

All other FM audio processors employ FM pre-emphasis prior to their multi-band limiting. This placement of the pre-emphasis function results in a more convenient design for the manufacturer. However, it becomes this approach in the processor which ends up sounding more dense and "packed up" in the higher audio frequencies. Most engineers and listeners have come to accept this sound as "the FM sound". This audio aberration is not a function of FM transmission, but a bad result of the design of traditional, multiband FM audio processors. Omnia takes a different approach. In the Omnia ONE, the necessary FM pre-emphasis is applied after the multi-band limiting. This technique requires difficult attention to both the limiting and clipping algorithms. However, the audible result is a cleaner, much more detailed high-end in the transmitted audio. This advantage of Omnia processing architecture is most easily noticed on musical instruments such as cymbals, castanets, trumpets, and other sounds with a lot of high-frequency energy. You will notice that these musical instruments sound "fake" through other audio processors, but they sound very real and natural through an Omnia audio processor.

### Separate multiband AGC and multiband limiter stages.

Most other audio processors take a shortcut in the execution of multiband AGC and limiting functions; they put both functions within the same audio processing "block". Omnia takes a more comprehensive approach. We designed the multiband AGC and multiband limiting blocks completely separate from each other. This allows us to give slightly different treatment as needed for the absolute cleanest and competitively loudest audio. For example, some bands of limiting are best served by feed-back servo control, while the higher bands are best served by feed-forward servo control. Our design, which breaks these multiband functions into individual processing blocks, allows for the absolute best treatment of each audio band.

### Omnia ONE's Powerful and Comprehensive Clipper

In any FM audio processor, the final audio clipper presents the largest hindrance or benefit to the loudness and clarity of the on-air sound for a given modulation level. The FM clipper section in the Omnia ONE, originally available only in big brother Omnia-6, includes two distinct sections: a bass-management clipper and a main clipper. The bass clipper is quite sophisticated in its own right, but the main clipper offers two different clipping styles and the ability to "balance" between them, if desired. This flexibility gives the curious or competitive user the ability to finely tune this most important function.

# Sensus Overview

Until now, digital signal processing has been a more precise numeric implementation of well-known analog methods. Even relatively recently designed digital audio processors couldn't veer too far from the comparatively simplistic concepts that analog dynamics processing had utilized...until now!

Extremely high power DSP chips have become available and at relatively low cost, and they make it possible to build smarter and more complex processing algorithms that were too difficult or impossible (or too expensive) to do in the past.

Running on a platform of the latest high power DSP chips, the Omnia ONE and our new Sensus® technology takes digital dynamics processing into a completely new frontier. Instead of the two-dimensional static processing architecture of the past, Sensus® enables the audio processor to modify its own architecture in real time and in response to ever-changing program content.

Simply stated, Sensus® has the ability to "sense" what must be done to a signal in order to best tailor it for output to a codec. As program content changes, it "rearranges the algorithms" to accomplish this goal. The uniqueness of the Sensus® technology makes it highly suitable not only for codec pre-conditioning (or provisioning), but also for a range of other highly specialized signal processing challenges. The following is a discussion of how Sensus® technology can be applied to a coded audio environment.

## Codec Provisioning

The codec is now a common denominator in the world of audio and broadcasting. Digital broadcasting (HDTV, HDRadioR, DAB, DRM), podcasting, webcasting, cellcasting, and downloadable music files all employ a form of codec-based data compression in order to minimize the bandwidth required to transmit audio data. The necessarily low bitrates utilized by these mediums presents a tough challenge for any audio processor used prior to a codec.

Traditional dynamics processors are designed to fulfill the requirements of a medium where the functions are generally static. That is, they're well suited to the rather simplistic peak control and bandwidth limiting methods required for analog broadcasting, as well as for the signal normalization techniques used in recording and mastering.

Audio codecs on the other hand are moving targets - each codec algorithm has its own set of artifacts. Not only does the sonic quality vary depending on the algorithm and bitrate used, but more importantly they vary in their ability to mask their own coding action. This is why we call it a 'moving target', and is why conventional audio processors fall short in a coded audio environment and can actually make coding artifacts worse due to their inability to adapt appropriately to the changing operation of the codec as the program content changes.

Prior art in audio dynamics processing could only address some of the challenges of provisioning audio for coding. This hurdle existed because the codec adapts to the incoming program (so as to generate the least amount of output data representing the input audio) causing the sonic artifacts generated by the process to continually change. Unless the audio processor can predict these changing characteristics of the codec, it can't possibly create output audio that is perfectly tailored for the coding process.

Conventional processors utilize rather simplistic high frequency limiters and fixed low pass filtering that does not change with the program material. When these less intelligent processors feed a codec the audio might sound acceptable one moment and offensive the next. Because they cannot "know" what the codec will do next, the result is over-compensated, dull and lifeless audio... audio that still contains objectionable codec-generated artifacts!

## Omnia ONE Multicast and HD® Radio

The advent of HD Radio$^{R}$ has introduced the capability to transmit multiple program streams, or "Multicast," within a single 96kbps digital broadcast data channel. To facilitate this, multicast relies on the use of codecs with comparatively low bitrates. A broadcaster can choose to transmit a number of multicast channels and select the bitrate for each one. However, the more multicast channels there are, the lower the bitrate each channel must have in order for them to all fit within thetotal available bandwidth.

In the very near future, many low bitrate multicast audio channels might exist. To achieve maximum sound quality, the kind that attracts and holds listeners, those channels will need specialized dynamics processing capable of creating great sound regardless of program content and bitrate. They will need Sensus®.